ENSC 427 - COMMUNICATION NETWORKS

Analysis of Quality of Service (QoS) for Video Conferencing in WiMAX Networks

Spring 2010 Final Project

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Abstract

WiMAX (Worldwide Interoperability for Microwave Access) certified telecommunication technology is also known as the IEEE 802.16 standard. This technology is revolutionizing the broadband wireless world by enabling the formation of a global wireless industry. WiMAX provides a theoretical maximum data rate of 75 Mbps on a single channel, and is designed to deliver next-generation high-speed mobile voice, video, and data services to a large geographical area.

WiMAX has the ability to support various types of applications, such as web browsing, e-mail, and file transfer. However, for applications that require real-time response, such as voice over IP (VoIP), streaming video and video conferencing, its Quality of Service (QoS) is inferior to traditional wired Internet due to packet loss over wireless mediums.

For this report, we use OPNET to analyze the QoS of WiMAX for video conferencing applications. Specifically, we analyze packet loss, transmitter power, distance, ARQ (Automatic Repeat Request), end-to-end delay, transmission mode, and buffer sizes in the video streams transported over the WiMAX network.

1.0 Introduction

1.1 Overview of this project

With the wide availability of high-speed broadband internet and the recent trend of computer manufacturers to include cameras and microphones in most laptop and desktop computers, the popularity of video conferencing has grown tremendously. This trend has been further aided by the growing number of computer applications that support video conferencing, such as Skype, Windows Live Messenger, and iChat. Businesses use video conferencing to conduct web meetings rather than paying the high cost of transporting employees to one physical location; educational institutes use it to implement distance-education programs; and the general public uses it to keep in touch with distant friends and family.

However, video conferencing is only available to customers with a high-speed, high-quality internet connection. Video conferencing requires at least 128 kbps (kilobits per second) of bandwidth both upstream and downstream for acceptable quality [6], which is only available with high-speed internet connections such as ADSL and cable. In addition, delay must be kept below approximately 100ms [6] in order to maintain flow in the conversation and prevent the participants from interrupting each other, which is difficult to attain in remote areas or long-range wireless connections such as satellite internet.

WiMAX (Worldwide Inter-operability for Microwave Access) is an emerging wireless last-mile internet technology that is promising to bring high-speed, high-quality wireless internet to large geographical areas. WiMAX provides a maximum transfer speed of 75Mbps per channel, and a maximum range of 50km (although, not both at the same time). Compared to ADSL, which has a maximum local-loop range of approximately 5km, WiMAX has a marked advantage in total area coverage. Also thanks to the large coverage area, users are able to stay connected to a high-speed internet connection while mobile, a major advantage over ADSL and cable. WiMAX simplifies the implementation of high-speed internet access to remote users, and is much cheaper to implement than wired systems.

This project focuses on the feasibility of using a WiMAX connection to provide last-mile internet connection for video conferencing. WiMAX is capable of providing the required bandwidth mentioned above over a large geographical area, however its QoS is, in general, lower than that of a wired connection [7]. Compared to wired connections, wireless connections have a much higher bit error rate (BER) due to the unpredictable nature of the transmission medium (air) and the obstacles between the transmitter and receiver (such as buildings and landforms). This necessitates re-transmissions, which leads to delay and jitter, and the need to buffer incoming data. This project will analyze the severity of these undesirable effects by simulating a WiMAX network in OPNET.

The structure of this project is as follows. Video conferencing and its necessary levels of quality of service (QoS) are explained in section 1.2. Details regarding WiMAX technology are provided in section 1.3. Section 2 describes our simulation in OPNET and discusses simulation results. Concluding remarks are provided in section 3. References are in section 4, lists of acronyms are in section 5, and OPNET simulation details are listed in the appendix.

1.2 Video conferencing and its necessary QoS

To set up a video conference, users need a camera, screen, microphone, speakers, software to process the audio and video, and a connection between the computers. Most modern computers are equipped with all the necessary hardware, and if they aren't then it is fairly cheap to obtain. There are many free video conferencing programs available, as mentioned in the introduction, as well as paid programs with more features, usually required for business conferences. The connection required between computers is usually an internet connection, but local area networks (LANs), cell-phone networks, and other proprietary networks are also used.

Video conferencing involves the transfer of audio and video between two users (point-to-point) or multiple users (multi-point). The video is encoded as a sequence of video frames, with frame rates ranging from 8 fps (frames per second) for low-bandwidth, low-quality video, to 30 fps or higher for high-quality video. The video is compressed using lossy compression codecs such as MPEG-4 or H.264 to save bandwidth. Compression ratios range from 1:10 up to 1:500, depending on the compression codec and video quality. This allows for high-quality video to be transmitted with as little as 256 kbps of bandwidth [6], rather than several megabits per second without compression. Lossless compression can also be used, but it has much lower compression ratios and the difference in quality between lossless and lossy video compression is not noticeable to most users.

Video conferences are conducted at various video resolutions. Higher resolutions are always preferred, but they require high data rates in order to maintain good quality video. Video resolution ranges from 128x96 for cell-phone video conferencing to 1920x1080 for important business or political video conferences.

Audio compression is also used to lower bandwidth requirements. Uncompressed mono CD-quality audio has a data rate of 723 kbps, but using audio compression codecs such as MP3 or AAC, high-quality mono audio can be transmitted with as little as 64kbps. Mono audio is used in most video conferences, as opposed to stereo, because having more than one audio channel for a single voice is unnecessary.

Table 1 shows a comparison of the audio and video quality used in video conferencing.

Table 1: Typical video and audio quality used in video conferencing [6]

Use	Video Resolution (pixels)	Video Data Rate (kbps)	Video Frame Rate (fps)	Audio Data Rate (kbps)
Cell phone	128x96	64	8 – 15	8 – 16
Low-quality PC	160x120	128	10	16
Medium-quality PC	320x240	256	15	32
High-quality PC	640x480	512	30	64
Business and political	640x480 +	512 +	30 +	64 +

As can be seen from Table 1, video conferencing demands fairly high bandwidth for good quality. It should be noted that the required bandwidth is needed in both data directions, and every additional user in a video conference necessitates more bandwidth. High-speed home internet connections have downlink speeds ranging from 256 kbps to 25 Mbps, and uplink speeds ranging from 128 kbps to 3 Mbps, with most connections being 3 Mbps downlink and 512 kbps uplink [3]. Therefore, video conferences on home-based internet connections are usually conducted at medium to low quality due to internet uplink connection speed limitations.

Video conferencing quality is very dependent on the internet connection used. Other than the obvious need for high bandwidth, the connection also requires low delay and packet loss in order to provide a high quality of experience (QoE). Delays of higher than approximately 100ms will be distracting and even disruptive to video conferences, causing multiple participants to accidentally speak at the same time. Packet loss can be handled in one of two ways. The lost packet can be retransmitted, which eliminates errors in a particular video or audio frame, but causes delay and jitter which are much more disruptive. Alternatively, the lost packet can be ignored, which produces a slight error in the audio or video stream. But, this error will only be momentary (as little as a few video frames or a few milliseconds of audio), and thus may not even be noticeable by the user. Therefore, it is preferable to ignore small amounts of packet loss in the interest of maintaining good delay and jitter performance. This is why video conferencing is referred to as a loss-tolerant, delay-sensitive service [1].

1.3 WiMAX Overview

WiMAX is a telecommunication technology that represents a family of IEEE 802.16 standards which focus on delivering fixed, nomadic, and mobile wireless internet access [1], [7]. It operates in the frequency range of 10 – 66 GHz with line of sight communications using a single carrier air interface [1], [9]. A recent addition to the 802.16 standard is a lower range of frequency bands which can operate from 2 – 11 GHz using one of three air interfaces: Single Carrier (SC), Orthogonal Frequency Division Multiplex (OFDM), and Orthogonal Frequency Division Multiple Access (OFDMA). OFDM and OFDMA enable carriers to increase their bandwidth and data capacity [1]. This increased efficiency is achieved by spacing subcarriers very closely together without interference because subcarriers are orthogonal to each other [1].

There are two types of WiMAX services, mobile and fixed. Mobile WiMAX enables users to access internet while traveling, and fixed WiMAX provides wireless internet access to fixed clients within a fixed radius. There are subscriber/client units which are available in both indoor and outdoor versions from several manufactures [7]. Usually, self-installed indoor units are convenient, but radio losses mean subscribers must be significantly closer to the WiMAX base station (BS) than with professionally-installed external units. The indoor-installed units require a higher infrastructure investment and operational cost due to the high number of BSs required to cover a given area [7]. However, indoor units are comparable in size to a cable modem or DSL modem. Outdoor units are roughly the size of a laptop PC, and their installation is comparable to the installation of a residential satellite dish.

Furthermore, WiMAX is an all-IP infrastructure deployed in a point-to-multi-point (PMP) topology, which is a specific and distinct type of multipoint connection, providing multiple paths from a single location to multiple locations [1]. It is able to achieve the required QoS by using a bandwidth request and granting scheme on the subscriber stations. This prevents the WiMAX BS from over-subscribing its available resources.

1.4 WiMAX Transmission Power

One of the challenges of designing for mobile WiMAX is its long range, since WiMAX networks typically achieve coverage of about 1 km per cell or base station [8]. To achieve long range, WiMAX networks require an optimized power profile from the base station to the components in the mobile device or any fixed workstation. For long range, high transmitter power is require but there is a limit. Designers and analyzers must find the optimal balance between high transmitter power and low power consumption to ensure robust links, high data rates, and good range for WiMAX services. WiMAX must simultaneously handle voice over Internet Protocol (VoIP), data, and video transmissions. Voice and video require managing the bandwidth and priority of transmission, which is a quality of service (QoS) component.

A typical WiMAX base station transmits at power levels of approximately +43dBm (20W) and a WiMAX mobile station (MS) typically transmits at +23 dBm (200mW) [8]. However, because WiMAX uses much higher modulation orders to achieve higher throughput, WiMAX requires a high SNR [8]. There is a large difference between downlink power (from the BS to the MS) and uplink power (from the MS to the BS), so mobile WiMAX networks are severely uplink limited.

WiMAX operates at high data rates and uses frequency division multiplexing (OFDM) with modulations from QPSK to 64-QAM, and is an all-IP-based network [8]. Mobile WiMAX networks employ a number of techniques to achieve longer range, including higher transmitter power, sub channelization, and adaptive modulation [8].

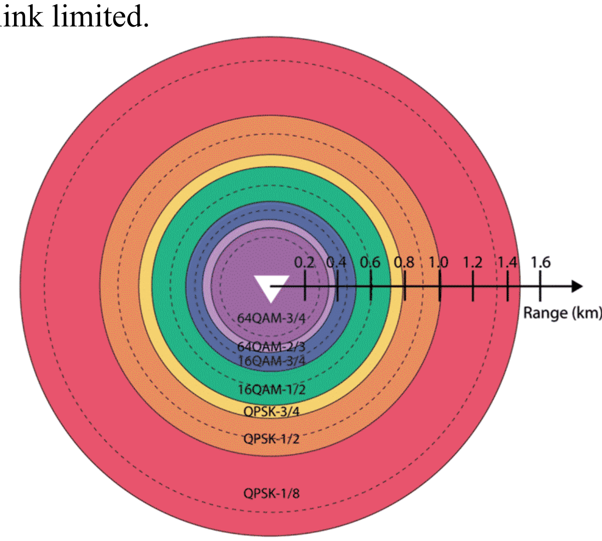


Figure 1: Achievable modulation versus distance with +23dBm transmit power [8]

In sub channelization, each MS concentrates its power over a subset of all available sub channels, and the other subcarriers are simultaneously made available to other users. In adaptive modulation, the MS transmits using lower-order modulation compared to the base station, to deal with the lower SNR that the mobile station has due to its lower transmitter power.

Figure 1 shows the modulation that is achievable as a function of distance from the BS. Mobile stations and workstations are often required to transmit using QPSK or 16-QAM signals, while the base station can usually use 64-QAM [8]. Because the SNR required to receive QPSK or 16-QAM is lower than 64-QAM, using a lower-order modulation allows the MS to communicate with the base station using less transmitter power.

The SNR required for QPSK-1/2 is 5 dB compared with 10.5 dB for 16-QAM-1/2 and 20 dB for 64-QAM-3/4 modulation [8]. If the mobile station transmits with QPSK, the base station can tolerate 5.5 dB more link loss than with 16-QAM.

Using a combination of sub channelization and adaptive modulation, a network operator can effectively balance the uplink and downlink budgets, and the network will operate bi-directionally.

2.0 OPNET Simulation and Discussion

For this report we simulated two WiMAX users using video conferencing applications in various scenarios. Our baseline scenario has the following parameters:

- 1km from the users to the BS
- 2W user transmitter power, 10W BS transmitter power
- Automatic Repeat Request (ARQ) disabled
- QPSK uplink modulation, 64-QAM downlink modulation
- 128KB downlink buffer, 64KB uplink buffer

We used the OPNET-defined Video Conferencing application for video, and the Voice applications for audio. The video parameters are 128kbps data rate and 30 FPS. The voice parameters are PCM Quality Sound with Silence suppressed, and G.711 encoding. We chose our baseline transmitter powers because they are typical WiMAX transmitter powers [8]. We initially chose 16-QAM for the uplink transmission modulation, but due to the low power of the users' transmitters, we found that the data loss was too high, so we switched to QPSK. The topology of our baseline scenario is shown in Figure 2. The user, application, profile, and WiMAX parameters are shown in the Appendix.

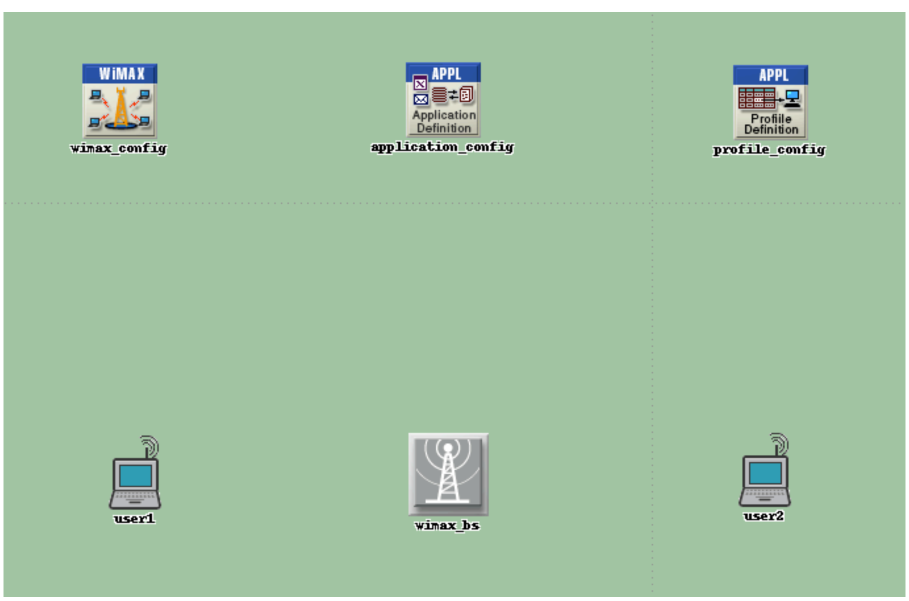
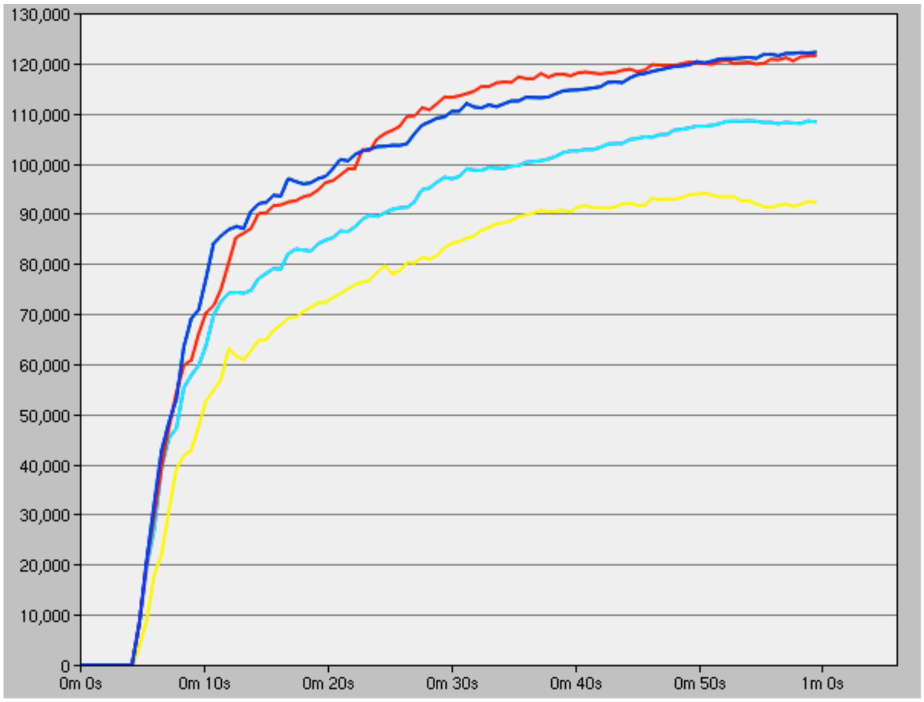


Figure 2: Baseline scenario topology

2.1 Transmitter Power

In our first scenario, we decreased the transmitter powers for the users and the BS to 0.2W and 1W, respectively. Figure 3 shows the results of this simulation. OPNET does not have statistics for data loss in the Video Conferencing or Voice applications, so instead the data rate of the sent and received data is displayed directly.



Blue: 10W BS, 2W user; data sent
Red: 1W BS, 0.2W user; data sent
Cyan: 10W BS, 2W user; data received
Yellow: 1W BS, 0.2W user; data received

Figure 3: Changing transmitter powers for the video application. Data rate in bps is shown on the y-axis and time on the x-axis. Values are averaged over time.

From this graph, it can be seen that the baseline scenario has a data loss of approximately 10%, and the decreased power scenario has a data loss of 25%. Therefore, the data loss rate is approximately logarithmically related to the transmitter power.

Both of these scenarios would be unacceptable for video conferencing. These scenarios would produce glitches in 25% or 10% of the video and audio, which would completely disrupt the conversation. Considering that 10W for the BS and 2W for the user are standard WiMAX transmitter powers, this suggests that other parameters need to be changed to produce an acceptable level of data loss. This idea will be explored further in later sections of this report.

Figure 4 shows the same comparison as in Figure 3, but this time the voice application is shown.

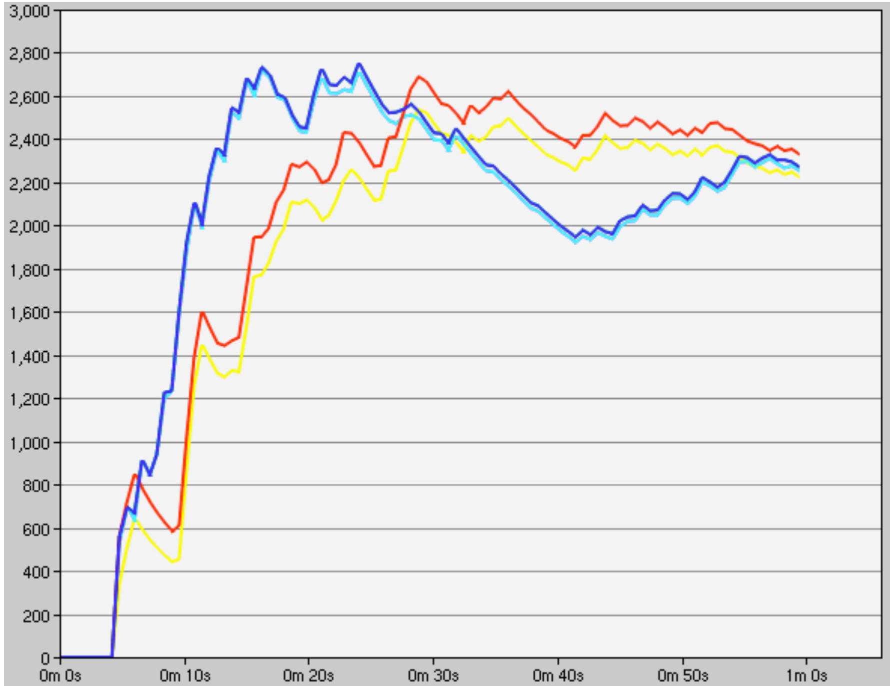


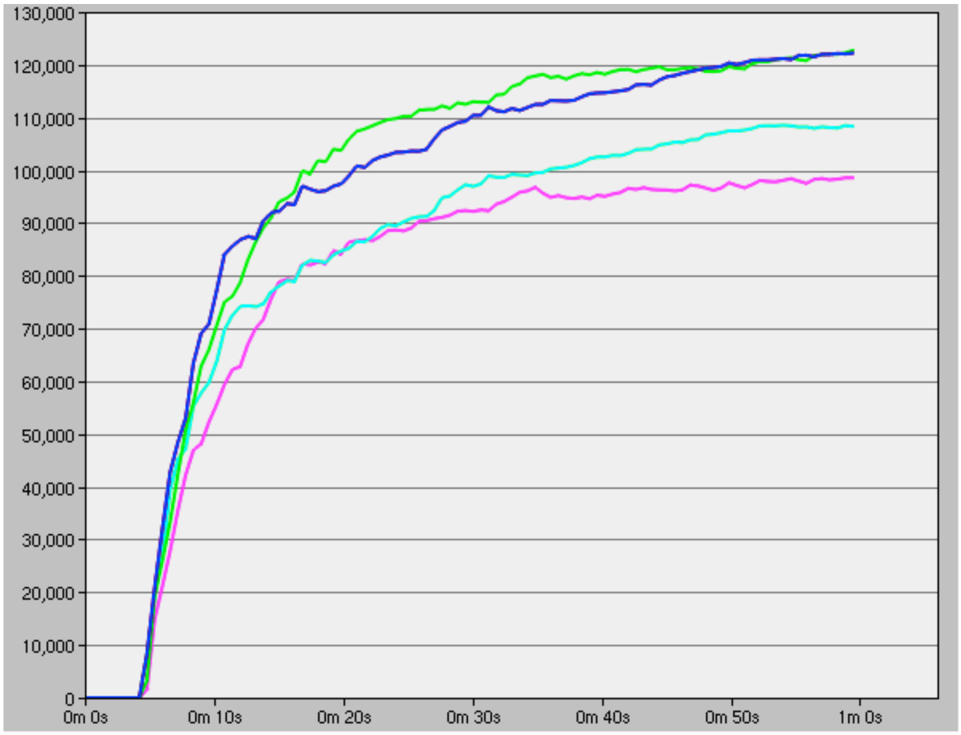
Figure 4: Changing transmitter powers for the voice application. Data rate in bps is shown on the y-axis and time on the x-axis. Values are averaged over time.

Results that are similar to the video application occur in this scenario, but the data loss is much less: approximately 1% for the baseline transmitter powers and 5% for the reduced transmitter powers. This is due to the fact that the Type of Service (ToS) setting in the Voice application attributes is set to Interactive Voice, whereas in the Video Conferencing application it is set to Interactive Multimedia, which has a lower priority than Interactive Voice. Therefore, setting the video ToS to Interactive Voice would be one method of decreasing the data loss. In OPNET's ToS settings, the user can also specify whether delay, throughput, or reliability, or a combination of the three, should be optimized in the network. However, care must be taken when setting these parameters because decreasing data loss will usually increase delay, which is also undesirable in a video conference.

The difference between video and voice results in all other simulations that we ran produced similar comparisons, and thus will not be shown.

2.2 Distance

For our second scenario, we moved the users to 1.5km from the BS, while keeping all other parameters the same as in the baseline scenario. The results of this scenario are shown in Figure 5.



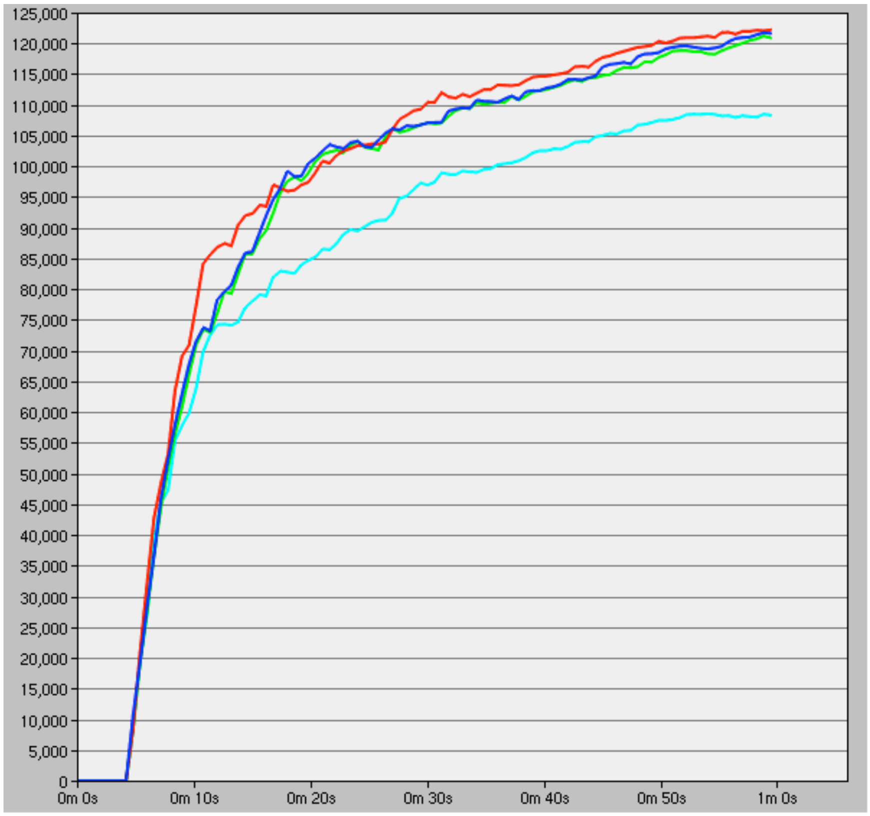
Blue: 1km to users; data sent
Green: 1.5km to users; data sent
Cyan: 1km to users; data received
Magenta: 1.5km to users; data received

Figure 5: Changing distance to users for the video application. Data rate in bps is shown on the y-axis and time on the x-axis. Values are averaged over time.

As can be seen, increasing the distance from the BS to the users by 50% approximately doubles the data loss. This scenario demonstrates the sensitivity of WiMAX to distance increases and the reason that the full transfer speed of a WiMAX connection cannot be used at far distances from the BS: because data loss will require frequent re-transmissions and hence a lower overall transfer speed.

2.3 Automatic Repeat Request (ARQ)

To decrease the data loss significantly, in this scenario we enabled ARQ while keeping all other parameters the same as the baseline scenario. The results are shown in Figure 6.



Blue: ARQ enabled; data sent
Red: ARQ disabled; data sent
Green: ARQ enabled; data receive
Cyan: ARQ disabled; data receive

Figure 6: ARQ enabled vs disabled. Data rate in bps is shown on the y-axis and time on the x-axis. Values are averaged over time.

This scenario shows that enabling ARQ will significantly decrease the data loss. Our baseline scenario has 10% data loss, whereas the ARQ-enabled scenario has slightly less than 1% data loss. Therefore, less than 1% of the video and audio in the video conference would have glitches, which would produce acceptable quality.

However, the disadvantage of enabling ARQ is that delay increases significantly. This is shown in Figure 7.

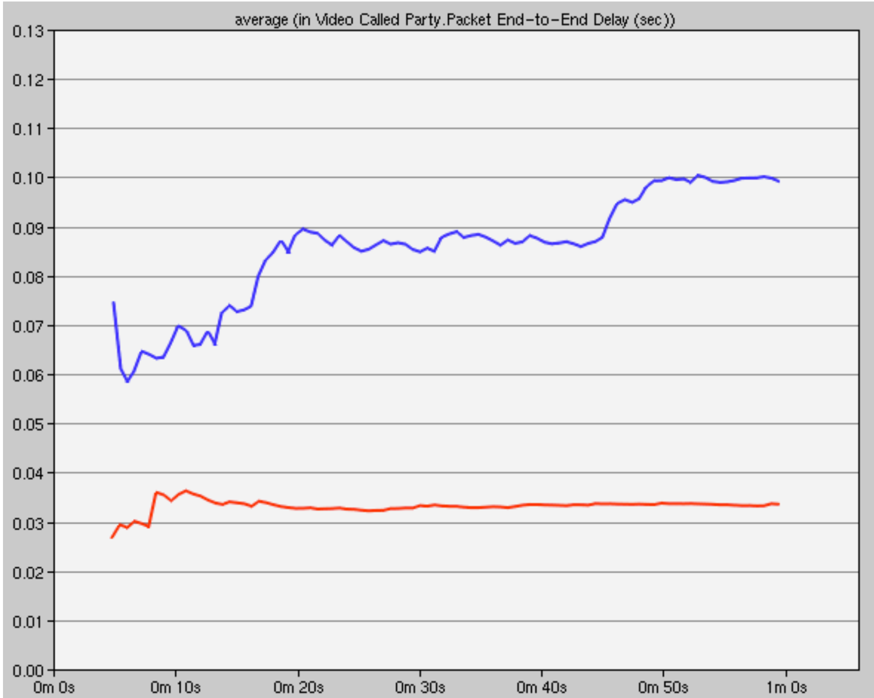


Figure 7: End-to-end delay in seconds. Values are averaged over time.

In this figure, the red line represents the baseline scenario (with ARQ disabled), and the blue line represents the scenario when ARQ is enabled with the default OPNET settings (see the Appendix for details). Enabling ARQ in our case increased the delay by a factor of three. In terms of video conferencing, enabling ARQ in our simulation will not cause much noticeable delay to the users because delay is only noticed by most users when it's above 100ms, which is the maximum delay that occurs in this scenario. However, when the users are more than 2km from each other, as is the case in our simulation, ARQ will either have to be disabled or its parameters will need to be changed to prevent the delay from going above 100ms.

2.4 Transmission Modulation

Next, we explored the effects of changing the transmission modulation in the uplink data direction while keeping all other parameters the same as the baseline scenario. Figure 8 shows the effects.

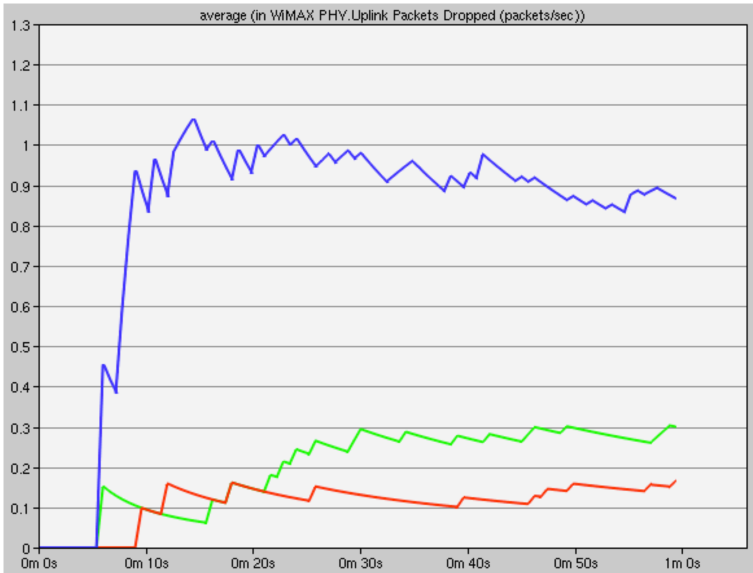


Figure 8: Changing transmission modulation in the uplink data direction. Values are packets dropped per second, and are averaged over time.

In this diagram, the blue line is the baseline scenario when QPSK is used, the red line is the scenario when 16-QAM is used, and the green line is the 64-QAM scenario. These results were unexpected because QPSK is theoretically the most robust modulation scheme, followed by 16-QAM, then 64-QAM. To investigate the cause of these results, were analyzed the queue size for each scenario, and this is shown in Figure 9.

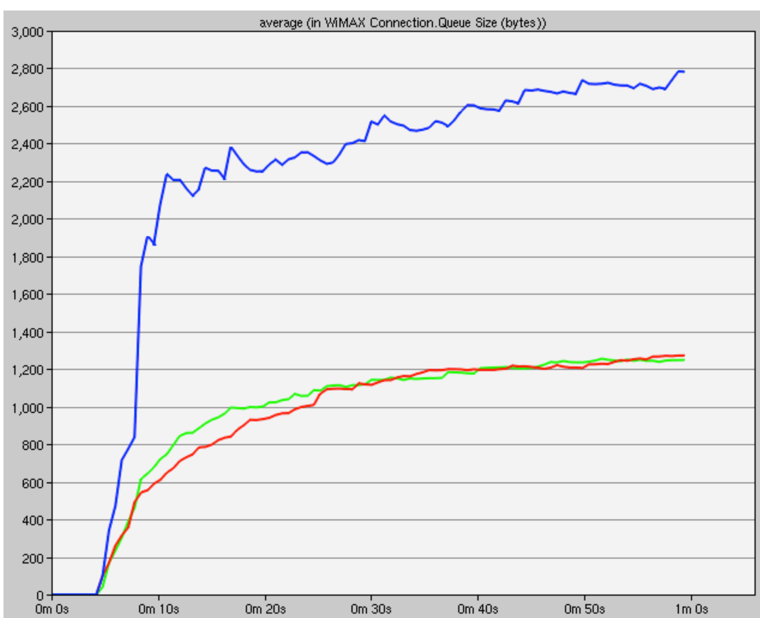


Figure 9: Queue size in bytes for each modulation scheme. Values are averaged over time.

In Figure 9, the blue line is for QPSK, the red line is for 16-QAM, and the green line is for 64-QAM. As can be seen, queue utilization for the 16-QAM and 64-QAM scenarios is almost the same, but the QPSK scenario has more than twice as much as queue utilization. We found that the reason for the high data loss of the QPSK scenario is that this modulation method was not transmitting the data fast enough. This would cause the queue to increase in size indefinitely, however the video conferencing application drops data after a specified amount of time, to reduce delay at the expense of lost data, so this is the reason for the data loss.

2.5 Buffer Sizes

In our final set of scenarios, we varied the sizes of both the uplink and downlink buffers in the users. The baseline scenario has buffer sizes of 128KB for downlink traffic and 64KB for uplink traffic. In one scenario we doubled these buffer sizes and in another we halved the sizes. To enable easier viewing of the results, we also changed the video to constant bit rate instead of exponentially-distributed variable bit-rate. Otherwise, all parameters are the same as in the baseline scenario. The results are shown in Figure 10.

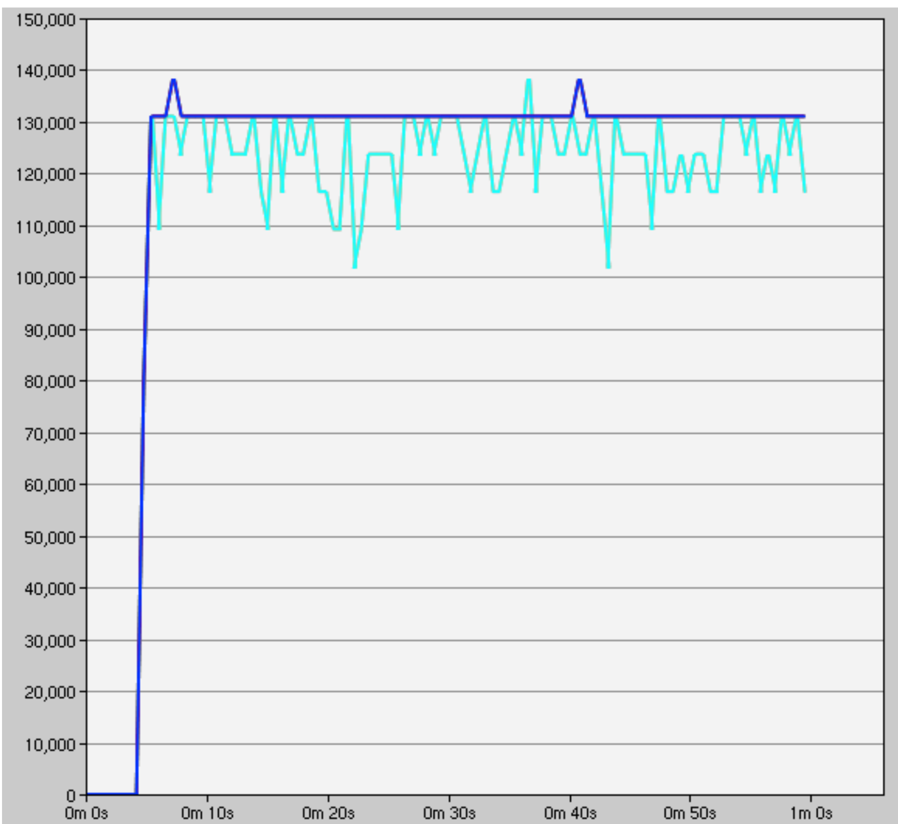


Figure 10: Changing buffer sizes. Values are shown as is (rather than time-averaged).

The blue line in this figure represents the constant data rate of the video data sent by user1, and the cyan line represents the data received by user2 in all cases of buffer sizes. Therefore, this graph shows that changing the buffer sizes does not affect the data loss. Further investigation revealed that this is due to the fact that the buffers were not being filled in any of the cases.

3.0 Future Work

To obtain further insight and better tuning of our simulation's performance, there are several aspects of the simulation that could use further work. Having larger distances between the users, such as 3000km, would simulate video conferencing situations that are more realistic because most video conferences are done between users who are otherwise unable to see each other in person. Having these long distances would introduce more data loss, which would require the use of fine-tuned ARQ settings to find a compromise between data loss and delay. The long distances and use of ARQ would probably also require fine-tuning of the buffer sizes. Varying the simulated terrain between the users and the BS would give insight into the effects of different physical landscapes. Finally, increasing the video data rate to high-quality video settings, such as 512kbps, would also be beneficial and would require re-tuning of all parameters.

4.0 Conclusion

In general, our OPNET results agree with theory. As expected, decreasing the transmitter powers increased data loss, increasing the distance from the users to the BS increased data loss, and enabling ARQ significantly decreased data loss at the expense of increased delay. We encountered unexpected results when changing the modulation scheme, but upon further investigation we found that the results were due to data being dropped because the data in the queue was not being transmitted fast enough in the QPSK scenario. Also, we unexpectedly found that changing the users' buffer sizes did not affect data loss, but later found that this was due to the fact that the buffers were not being filled in any of the scenarios.

The only scenario in our simulation that produced nearly acceptable results for video conferencing was the one in which ARQ is enabled. Even this scenario had fairly high data loss (nearly 1%), which means that there is further improvement possible in our simulation. A combination of the measures mentioned in the Future Work section would greatly increase the QoS of video conferencing, and should be considered for future simulations.

This project presented several challenges, such as generating a working WiMAX network in OPNET, setting attributes to make the network realistic, and determining what caused the unexpected results we obtained. Also, learning about the technical aspects of WiMAX and video conferencing was insightful and enabled us to determine a good set of scenarios to run. Obtaining statistics in OPNET was fairly non-trivial, but did require considerable patience due to the lengthy simulation times of OPNET and the slow Sun computers that OPNET is installed on.

5.0 References

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6.0 – List of Acronyms

ARQ	Automatic Repeat Request
BS	Base Station
BER	Bit Error Rate
FPS	Frames Per Second
kbps	kilobits per second (as opposed to kBps, which is kilo*bytes* per second)
LAN	Local Area Network
OFDMA	Orthogonal Frequency Division Multiple Access
OFDM	Orthogonal Frequency Division Multiplex
PMP	Point-to-Multi-Point
QAM	Quadrature Amplitude Modulation
QoE	Quality of Experience
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
SC	Single Carrier
ToS	Type of Service
VoIP	Voice over IP
WiMAX	Worldwide Inter-operability for Microwave Access

7.0 – Appendix: OPNET code listing

The following are screenshots of the attribute windows for the various settings in our OPNET simulation baseline scenario. We did not modify any code, therefore there is no code shown here.

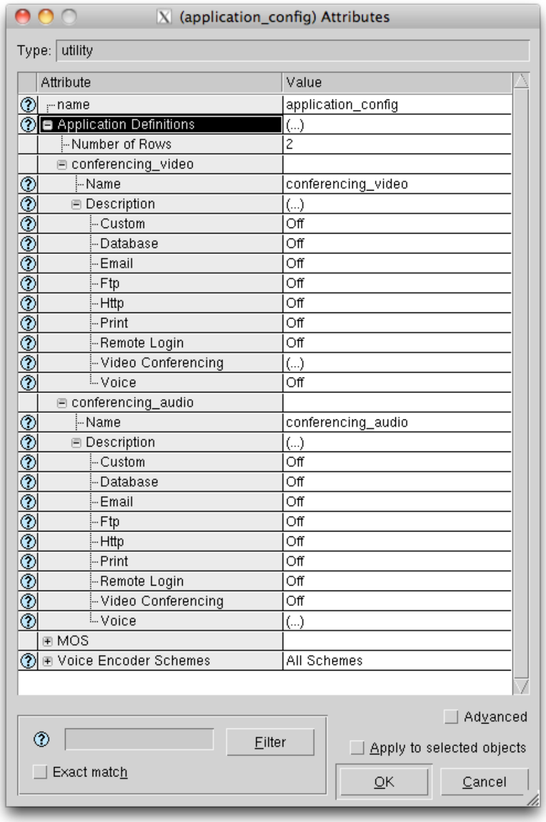


Figure 11: Application configuration

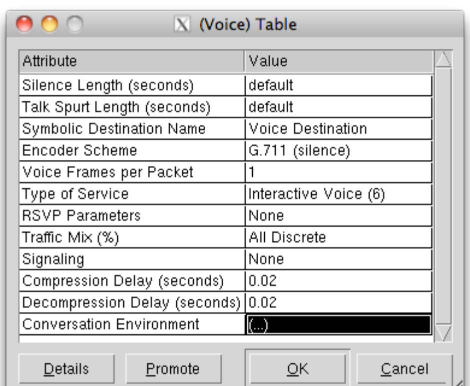


Figure 12: Voice application settings. This application is a modification of the OPNET-defined PCM Quality and Silence Suppressed application

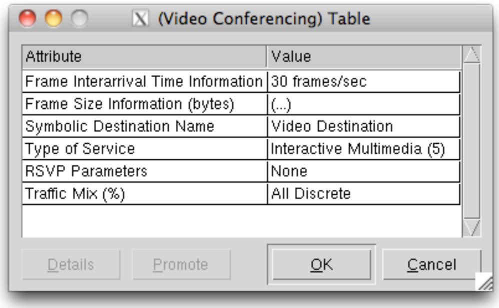


Figure 13: Video conferencing application settings

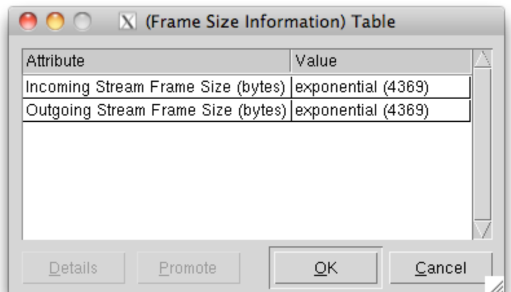


Figure 14: Video conferencing frame size

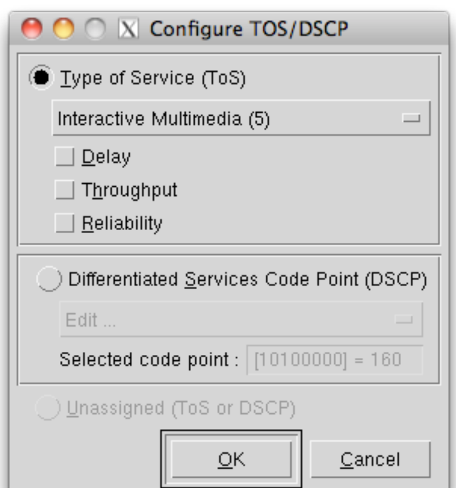
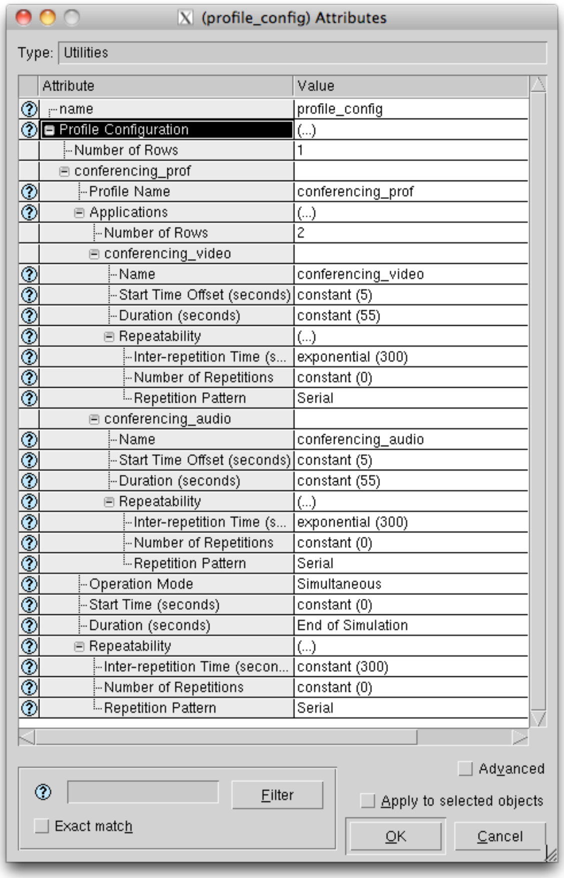


Figure 15: Video conferencing Type of Service (ToS)



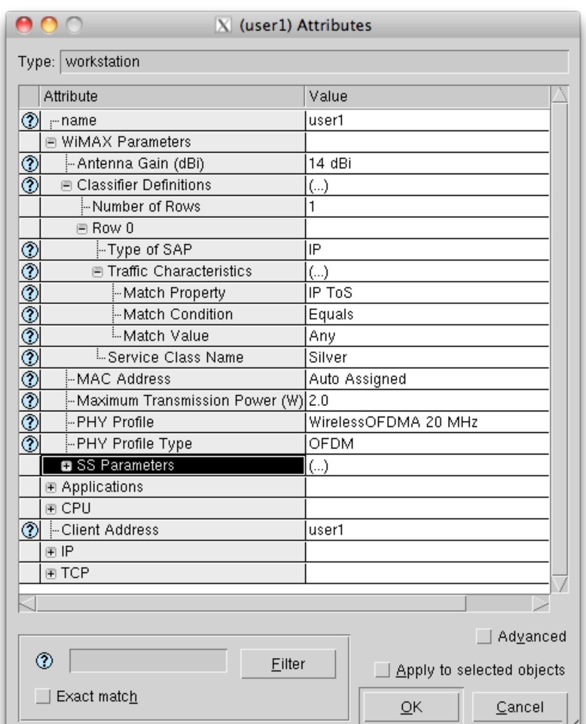
(profile_config) Attributes

Type: Utilities

Attribute	Value
name	profile_config
Profile Configuration	(...)
Number of Rows	1
conferencing_prof	
Profile Name	conferencing_prof
Applications	(...)
Number of Rows	2
conferencing_video	
Name	conferencing_video
Start Time Offset (seconds)	constant (5)
Duration (seconds)	constant (55)
Repeatability	(...)
Inter-repetition Time (s...	exponential (300)
Number of Repetitions	constant (0)
Repetition Pattern	Serial
conferencing_audio	
Name	conferencing_audio
Start Time Offset (seconds)	constant (5)
Duration (seconds)	constant (55)
Repeatability	(...)
Inter-repetition Time (s...	exponential (300)
Number of Repetitions	constant (0)
Repetition Pattern	Serial
Operation Mode	Simultaneous
Start Time (seconds)	constant (0)
Duration (seconds)	End of Simulation
Repeatability	(...)
Inter-repetition Time (secon...	constant (300)
Number of Repetitions	constant (0)
Repetition Pattern	Serial

Advanced
Filter
Apply to selected objects
Exact match
OK
Cancel

Figure 16: Profile configuration



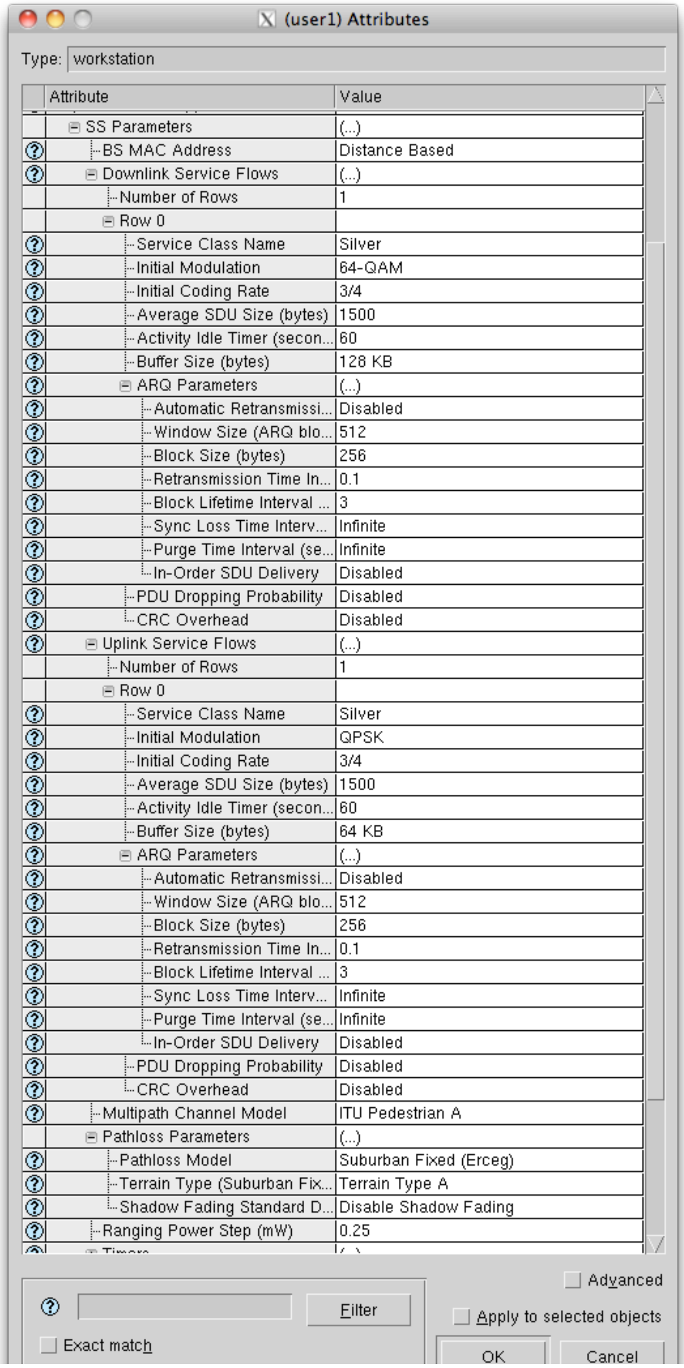
(user1) Attributes

Type: workstation

Attribute	Value
name	user1
WiMAX Parameters	
Antenna Gain (dBi)	14 dBi
Classifier Definitions	(...)
Number of Rows	1
Row 0	
Type of SAP	IP
Traffic Characteristics	(...)
Match Property	IP ToS
Match Condition	Equals
Match Value	Any
Service Class Name	Silver
MAC Address	Auto Assigned
Maximum Transmission Power (W)	2.0
PHY Profile	WirelessOFDMA 20 MHz
PHY Profile Type	OFDM
SS Parameters	(...)
Applications	
CPU	
Client Address	user1
IP	
TCP	

Advanced
Filter
Apply to selected objects
Exact match
OK
Cancel

Figure 17: User WiMAX parameters, part 1



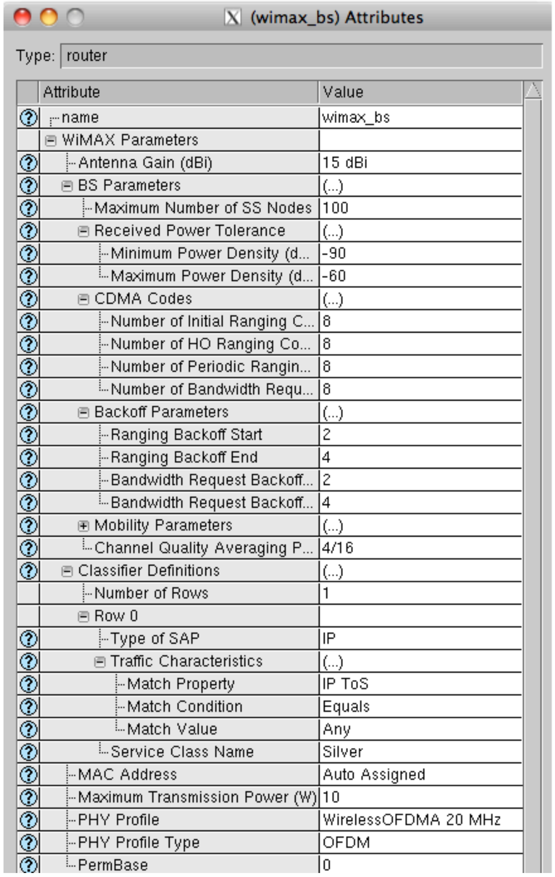
(user1) Attributes

Type: workstation

Attribute	Value
SS Parameters	(...)
BS MAC Address	Distance Based
Downlink Service Flows	(...)
Number of Rows	1
Row 0	
Service Class Name	Silver
Initial Modulation	64-QAM
Initial Coding Rate	3/4
Average SDU Size (bytes)	1500
Activity Idle Timer (secon...	60
Buffer Size (bytes)	128 KB
ARQ Parameters	(...)
Automatic Retransmissi...	Disabled
Window Size (ARQ blo...	512
Block Size (bytes)	256
Retransmission Time In...	0.1
Block Lifetime Interval ...	3
Sync Loss Time Interv...	Infinite
Purge Time Interval (se...	Infinite
In-Order SDU Delivery	Disabled
PDU Dropping Probability	Disabled
CRC Overhead	Disabled
Uplink Service Flows	(...)
Number of Rows	1
Row 0	
Service Class Name	Silver
Initial Modulation	QPSK
Initial Coding Rate	3/4
Average SDU Size (bytes)	1500
Activity Idle Timer (secon...	60
Buffer Size (bytes)	64 KB
ARQ Parameters	(...)
Automatic Retransmissi...	Disabled
Window Size (ARQ blo...	512
Block Size (bytes)	256
Retransmission Time In...	0.1
Block Lifetime Interval ...	3
Sync Loss Time Interv...	Infinite
Purge Time Interval (se...	Infinite
In-Order SDU Delivery	Disabled
PDU Dropping Probability	Disabled
CRC Overhead	Disabled
Multipath Channel Model	ITU Pedestrian A
Pathloss Parameters	(...)
Pathloss Model	Suburban Fixed (Erceg)
Terrain Type (Suburban Fix...	Terrain Type A
Shadow Fading Standard D...	Disable Shadow Fading
Ranging Power Step (mW)	0.25

Advanced
Filter
Apply to selected objects
Exact match
OK
Cancel

Figure 18: User WiMAX parameters, part 2

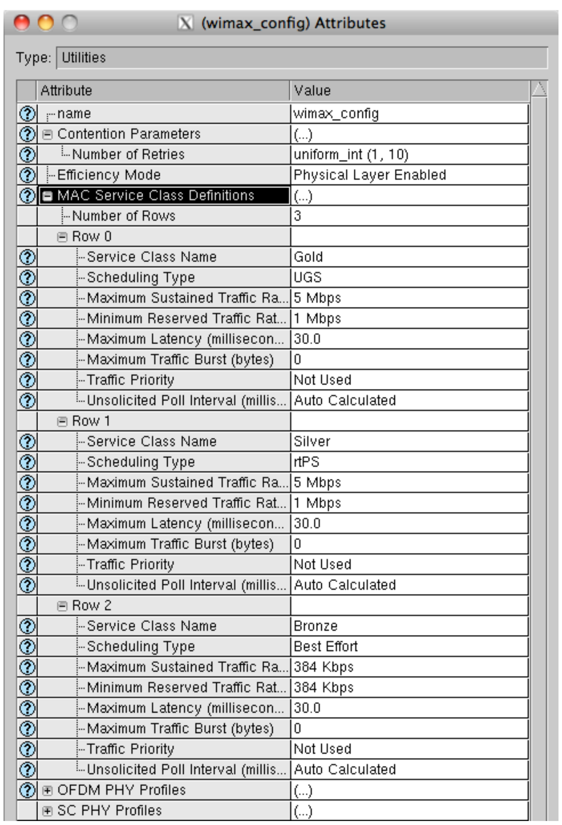


(wimax_bs) Attributes

Type: router

Attribute	Value
name	wimax_bs
WiMAX Parameters	
Antenna Gain (dBi)	15 dBi
BS Parameters	(...)
Maximum Number of SS Nodes	100
Received Power Tolerance	(...)
Minimum Power Density (d...	-90
Maximum Power Density (d...	-60
CDMA Codes	(...)
Number of Initial Ranging C...	8
Number of HO Ranging Co...	8
Number of Periodic Rangin...	8
Number of Bandwidth Requ...	8
Backoff Parameters	(...)
Ranging Backoff Start	2
Ranging Backoff End	4
Bandwidth Request Backoff...	2
Bandwidth Request Backoff...	4
Mobility Parameters	(...)
Channel Quality Averaging P...	4/16
Classifier Definitions	(...)
Number of Rows	1
Row 0	
Type of SAP	IP
Traffic Characteristics	(...)
Match Property	IP ToS
Match Condition	Equals
Match Value	Any
Service Class Name	Silver
MAC Address	Auto Assigned
Maximum Transmission Power (W)	10
PHY Profile	WirelessOFDMA 20 MHz
PHY Profile Type	OFDM
PermBase	0

Figure 19: BS WiMAX parameters

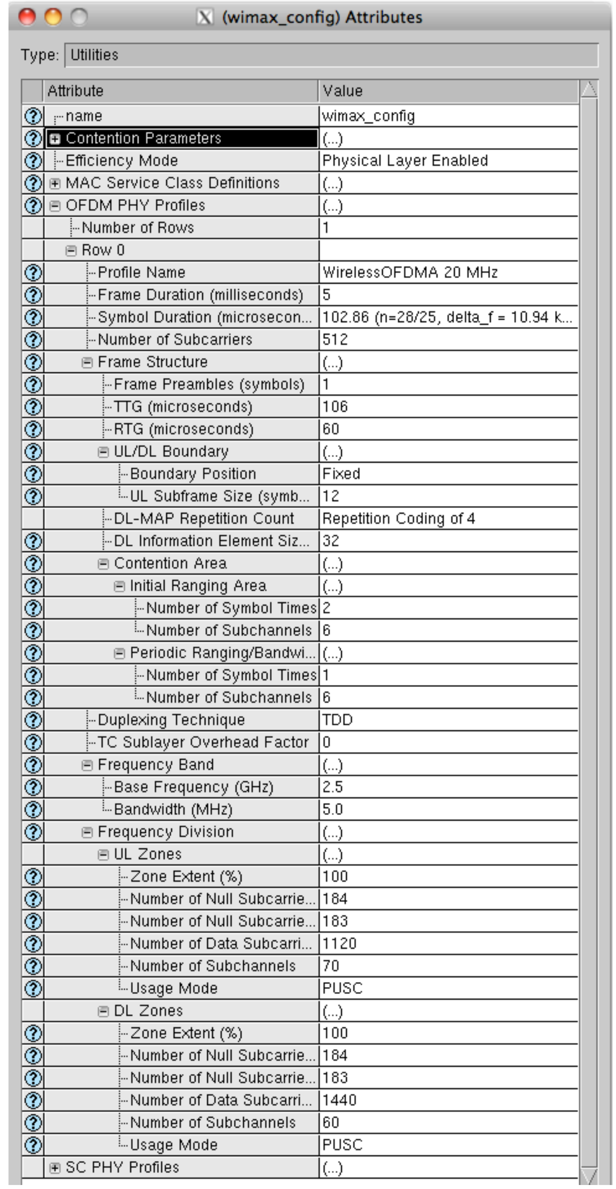


(wimax_config) Attributes

Type: Utilities

Attribute	Value
name	wimax_config
Contention Parameters	(...)
Number of Retries	uniform_int (1, 10)
Efficiency Mode	Physical Layer Enabled
MAC Service Class Definitions	(...)
Number of Rows	3
Row 0	
Service Class Name	Gold
Scheduling Type	UGS
Maximum Sustained Traffic Ra...	5 Mbps
Minimum Reserved Traffic Rat...	1 Mbps
Maximum Latency (millisecon...	30.0
Maximum Traffic Burst (bytes)	0
Traffic Priority	Not Used
Unsolicited Poll Interval (millis...	Auto Calculated
Row 1	
Service Class Name	Silver
Scheduling Type	rtPS
Maximum Sustained Traffic Ra...	5 Mbps
Minimum Reserved Traffic Rat...	1 Mbps
Maximum Latency (millisecon...	30.0
Maximum Traffic Burst (bytes)	0
Traffic Priority	Not Used
Unsolicited Poll Interval (millis...	Auto Calculated
Row 2	
Service Class Name	Bronze
Scheduling Type	Best Effort
Maximum Sustained Traffic Ra...	384 Kbps
Minimum Reserved Traffic Rat...	384 Kbps
Maximum Latency (millisecon...	30.0
Maximum Traffic Burst (bytes)	0
Traffic Priority	Not Used
Unsolicited Poll Interval (millis...	Auto Calculated
OFDM PHY Profiles	(...)
SC PHY Profiles	(...)

Figure 20: WiMAX configuration, part 1



(wimax_config) Attributes

Type: Utilities

Attribute	Value
name	wimax_config
Contention Parameters	(...)
Efficiency Mode	Physical Layer Enabled
MAC Service Class Definitions	(...)
OFDM PHY Profiles	(...)
Number of Rows	1
Row 0	
Profile Name	WirelessOFDMA 20 MHz
Frame Duration (milliseconds)	5
Symbol Duration (microsecon...	102.86 (n=28/25, delta_f = 10.94 k...
Number of Subcarriers	512
Frame Structure	(...)
Frame Preambles (symbols)	1
TTG (microseconds)	106
RTG (microseconds)	60
UL/DL Boundary	(...)
Boundary Position	Fixed
UL Subframe Size (symb...	12
DL-MAP Repetition Count	Repetition Coding of 4
DL Information Element Siz...	32
Contention Area	(...)
Initial Ranging Area	(...)
Number of Symbol Times	2
Number of Subchannels	6
Periodic Ranging/Bandwi...	(...)
Number of Symbol Times	1
Number of Subchannels	6
Duplexing Technique	TDD
TC Sublayer Overhead Factor	0
Frequency Band	(...)
Base Frequency (GHz)	2.5
Bandwidth (MHz)	5.0
Frequency Division	(...)
UL Zones	(...)
Zone Extent (%)	100
Number of Null Subcarrie...	184
Number of Null Subcarrie...	183
Number of Data Subcarri...	1120
Number of Subchannels	70
Usage Mode	PUSC
DL Zones	(...)
Zone Extent (%)	100
Number of Null Subcarrie...	184
Number of Null Subcarrie...	183
Number of Data Subcarri...	1440
Number of Subchannels	60
Usage Mode	PUSC
SC PHY Profiles	(...)

Figure 21: WiMAX configuration, part 2